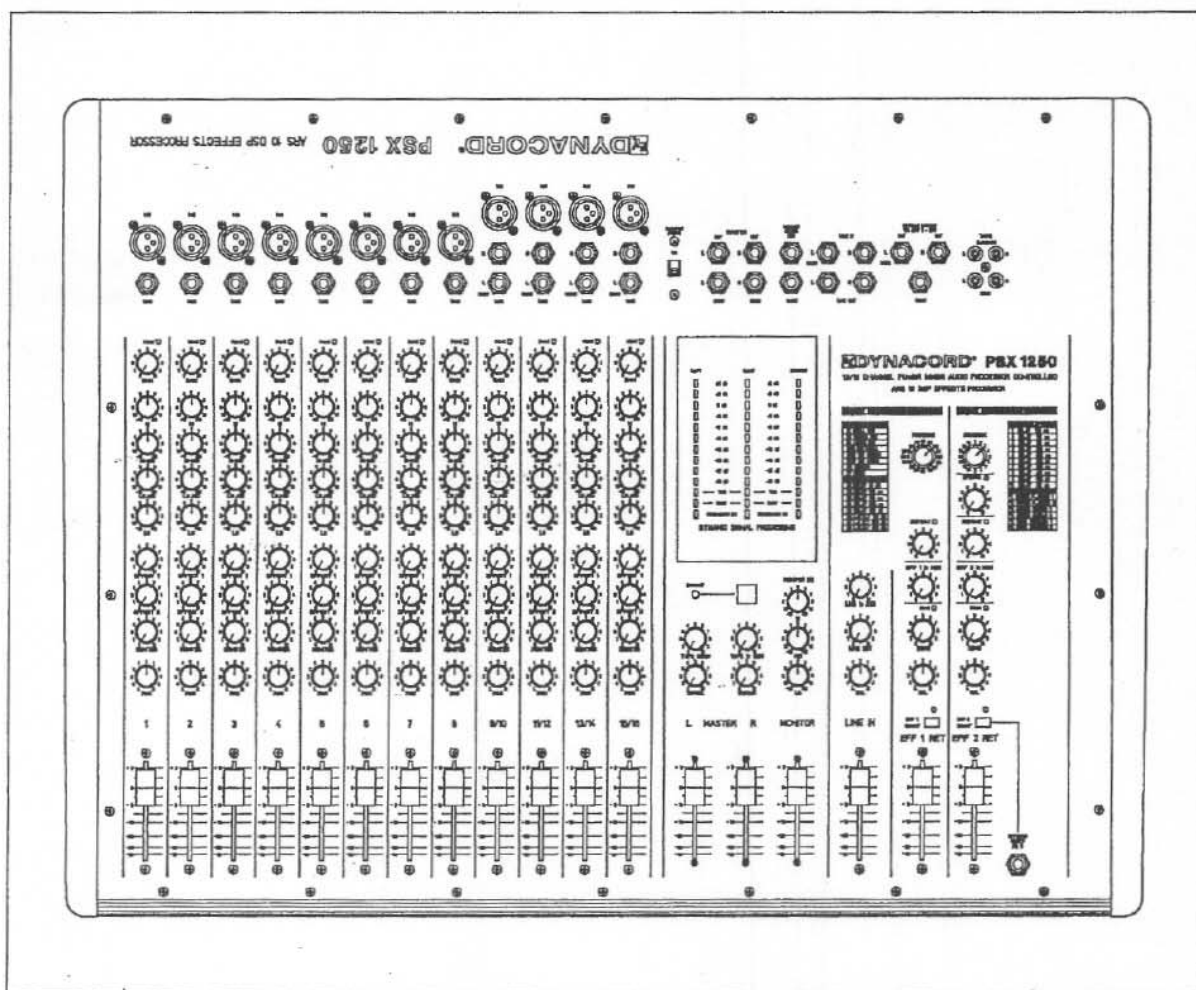


EV[®] DYNACORD[®]

USER MANUAL



PSX 1250

12/16 CHANNEL POWER MIXER

IMPORTANT NOTES

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IMPORTANT NOTES

CAUTION: The unit must be protected from humidity to prevent the risk of fire or electric shock.

1. Make sure that you have the correct mains voltage. Only operate with the unit with the mains voltage marked on the rear panel.
2. Ensure that no objects (in particular metal objects) are inserted into the unit. This could cause severe electric shock or give rise to malfunction.
3. If the unit is suddenly moved from a cold place to a warm place, e.g. transporting from outside into a heated room, condensation will occur. In this case you should not switch the unit on until it has warmed up to room temperature.
4. In the event of water or any other fluid being accidentally spilt on the unit, switch the unit off immediately and send it to a qualified service workshop for inspection.
5. Always ensure that the unit is well ventilated and never exposed to direct sunlight.
6. Do not use sprays to clean the unit as these may cause damage to it and could suddenly ignite.

PSX 1250 - General Overview

- 12 Input channels = 8 x Mic / Line + 4 Mic / Line Stereo
- 4-band equalizer in each input channel
- Two separate digital 16 bit stereo effect units
- 1 digital 16 bit stereo effect unit with 6 reverb 8 delay programs and 2 special effects
- 1 digital 16 bit stereo effect unit with 16 delay, flanging, chorus and doubling programs
- 48 V Phantom Power
- 2 effect paths
- 1 AUX/Monitor path with 3-band equalizer and separate PCA processor power amp
- Controllable Line In and Line Out path
- Controllable Tape Record and Tape Playback path
- Big 3-way LED Level Meter
- 2 x 250 Watts (RMS/4 Ohms) PCA processor power amplifiers for master
- 1 x 200 Watts (RMS/2 Ohms) PCA processor power amplifier for monitor

The mixer of the PSX compact power mixers is equipped with a wide range of features. The stereo input channels allow the connection of stereo program sources like keyboards, drum machines, tape recorders and additional mixing desks. Thus a lot more sound sources can be connected than with conventional power mixers. The stereo input channels can also be used as regular mic input channels. All mic inputs are electronically balanced and equipped with XLR sockets. 48 volt Phantom power can be switched to the mic sockets.

The effect section is equipped with 2 digital 16 bit stereo reverb/delay/ special effect modules. One effect unit generates different, extremely natural sounding stereo reverb programs, special stereo programs which combine the reverb programs with additional echoes, and a stereo chorus program especially structured for vocals, brass and woodwinds. In addition 8 different delay and echo programs with excellent quality are available from the ARS 10 DSP effects board. The number of echo repeats is front-panel controllable.

The other digital effect unit generates 16 different delay, flanging, chorus and doubling programs.

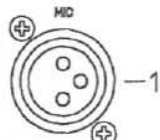
The master section is equipped very comprehensively with separate control functions, e.g. for tape playback and record, connection for an additional external stereo effect unit, separately controllable AUX/LINE output/input and big LED level meter for the power amplifiers.

The PCA processor power amplifiers are designed in advanced Hi-Power MOS technology. The frequency and phase response of the loudspeaker cabinets connected is linearized in the low-frequency region with a 2nd order shelving eq. The corner frequencies of the 2nd order shelving eq have been designed to match with the characteristics of modern high-efficiency loudspeaker cabinets. A built-in fast acting limiter prevents excessive overdrive.

The power outputs of the power amps are equipped with speakon adapters. These connectors were developed especially for the peak performance of modern power amps and guarantee a safe and loss-free connection of loudspeaker cables with the greatest possible cable cross-sectional area.

INPUT / MONO

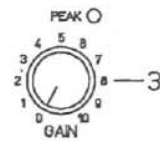
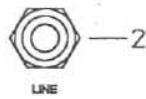
INPUT / MONO



1. Mic

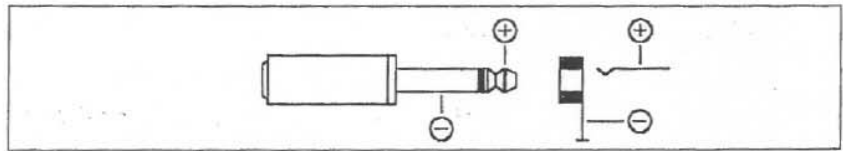
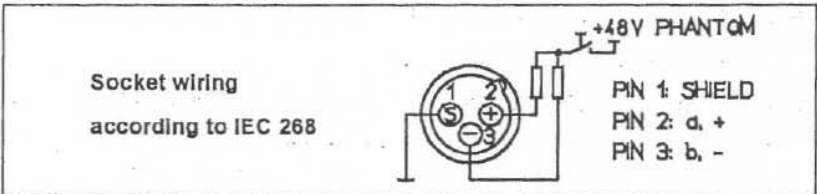
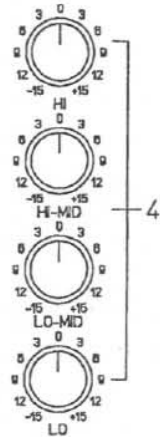
Electronically balanced XLR inputs 1-8 for connection of low-impedance microphones. Also phantom-powered microphones (condenser microphones) can be connected to these sockets.

For further information see: **63. Phantom Power.**

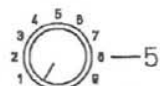


2. LINE

Unbalanced inputs 1-8 for instruments and other high level signal sources. The insertion of a plug into this socket will switch off the XLR input.



Please make sure that the respective channel fader or at least both master faders are closed before connecting signal sources, to protect your audience from annoying click noise.



3. GAIN + PEAK LED

Control for adjusting the input sensitivity between -56 dBV (1.5 mV) and -20 dBV (95 mV).

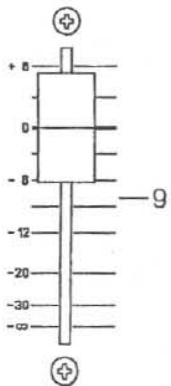
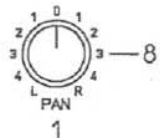


This control should be adjusted so that the PEAK LED lights up only briefly at peak levels. This will result in an optimum S/N ratio. If the PEAK LED lights up, the available headroom is 10 dB before the input signal is distorted audibly. Please note that the sound corrections with the EQ controls influence the input signal level as well. Check the correct setting of the GAIN control again after sound adjustment.



4. EQUALIZING

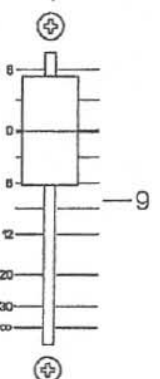
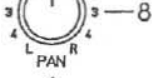
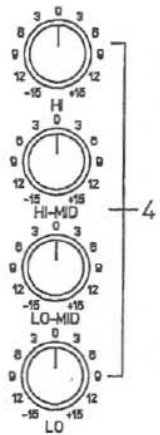
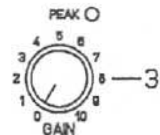
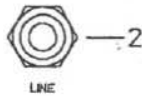
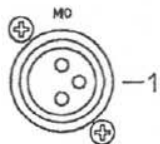
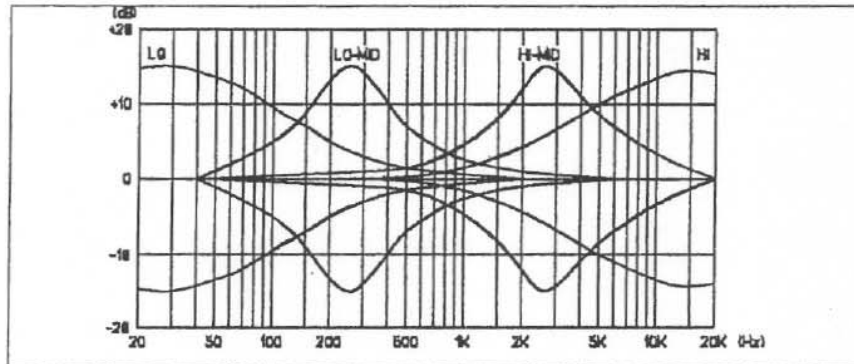
The EQ section allows for a very comprehensive and effective control of the input signal for the different frequency regions. Turning the control to the right increases the respective frequency region. Turning left will decrease the respective frequency region.



When the sound is to be adjusted, you should always start in a neutral position, i.e. all controls are in central position. If possible, do not use extreme control positions; normally a small sound correction is enough and delivers the best sound quality. Take as orientation the naturalness of the reproduction and depend on your musically trained ear for sound checking. You can effectively avoid acoustic feedback by operating the mid controls (MID) gently. Avoid too much gain especially in these frequency regions; a light lowering allows an amplification of microphone signals with little danger of feedback.

4. EQUALIZING (continuation of page 4)

Control	Variation	Frequency	Type
HI	+/-15 dB	15 kHz	shelving
HI-MID	+/-10 dB	2.5 kHz	boost/cut
LO-MID	+/-10 dB	250 Hz	boost/cut
LO	+/-15 dB	50 Hz	shelving



5. EFFECT 1

Control for adjusting the EFFECT 1 level. This control is electrically arranged after (post fader) the channel fader (9), so that the signal level depends on the position of the channel fader. Please control the send signal to the integrated effect module carefully. The PEAK indication of the effect module may only light up briefly at dynamic signal peaks. If the LED is lit continuously, the unit is being overdriven.

The EFFECT 1 path can be used to send a signal to a separate external effect unit or to drive a separate monitor power amp.

For further information see: **36-44, EFFECT 1**

6. EFFECT 2

Control for sending a signal to the built-in digital effect unit (reverb/echo). This control is also arranged after (post fader) the channel fader (9); the effect signal level depends on the position of the channel fader as well.

Please control the send signal to the integrated effect module carefully. The PEAK indication of the effect module (49) may only light up briefly at dynamic signal peaks. If the LED is lit continuously, the unit is being overdriven.

For further information see: **45-53, EFFECT 2**

7. MONITOR

Control for driving a monitor power amp. This control does not depend on the position of the channel fader (9).

8. PAN

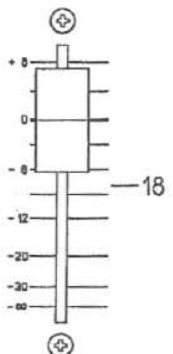
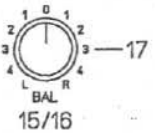
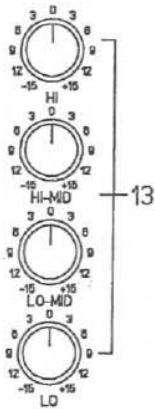
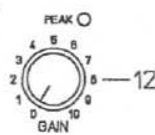
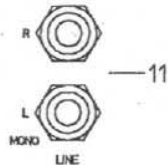
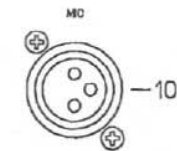
This control determines the stereo position of the input signal. In central position the signal is divided into equal parts on both master channels left and right.

9. CHANNEL FADER

The channel fader is used to adjust the volume of the single channel and the volume balance between the individual channels.

Try to adjust the fader near the 0 dB position. This will enable you to adjust the volume with sufficient control displacement even if you have great level differences between the different input channels. The master volume of the complete unit is controlled by the master faders.

INPUT / STEREO



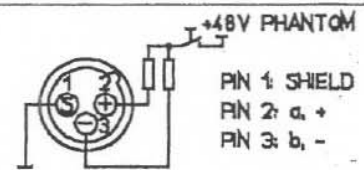
INPUT / STEREO

The input channels 9 to 12 are designed completely in stereo. You can connect all types of stereo signal sources (e.g. drums, synthesizers, samplers or submixers).

11. Mic

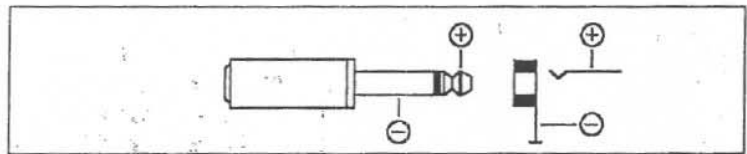
The stereo input channels can, of course, also be operated in mono. With low impedance microphones you should use the XLR socket MIC and with higher level signal sources the jack LINE L/MONO (11).

Socket wiring according to IEC 268



12. LINE

Unbalanced inputs L + R for instruments and other high level signal sources. The insertion of a plug into these sockets will switch off the XLR input.



Please make sure that the respective channel fader or at least both master faders are closed before connecting signal sources, to protect your audience from annoying click noise.

13. GAIN + PEAK LED

Control for adjusting the input sensitivity between -56 dBV (1.5 mV) and -20 dBu (100 mV).

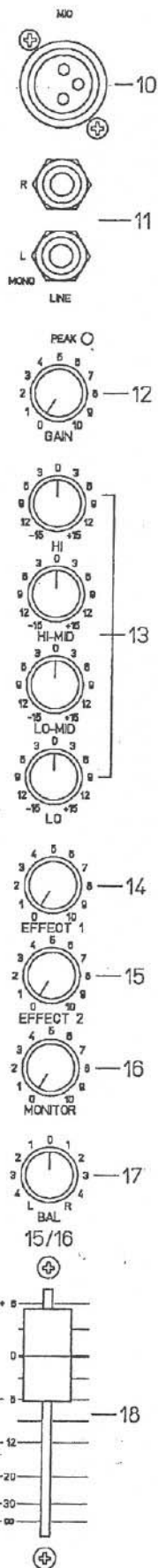
This control should be adjusted so that the PEAK LED only lights up briefly at signal peaks. This will result in an optimum S/N ratio. If the PEAK LED lights up, the available headroom is 10 dB before the input signal is distorted audibly. Please note that the sound corrections with the EQ controls influence the input signal level as well. Check the correct setting of the GAIN control again after adjusting the sound.

14. EQUALIZING

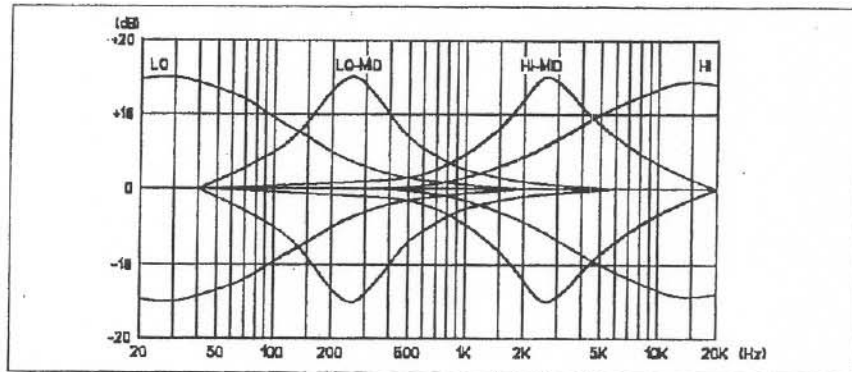
The EQ section allows for a very comprehensive and effective control of the input signal for the different frequency regions. Turning the control to the right increases the respective frequency region. Turning left will decrease the respective frequency region.

When adjusting you should always start at a neutral position, i.e. all controls are in central position. If possible, do not use extreme control positions; normally a small sound correction is enough and delivers the best sound quality. Take as orientation the naturalness of the reproduction and depend on your musical trained ear for sound checking. You can effectively avoid acoustical feedback by operating of mid controls (MID) gently. Avoid too much gain especially in these frequency regions; a slight lowering allows an amplification of microphone signals with little danger of feedback.

13. EQUALIZING (continuation of page 6)



Control	Variation	Frequency	Type
HI	+/-15 dB	15 kHz	shelving
HI-MID	+/-10 dB	2.5 kHz	boost/cut
LO-MID	+/-10 dB	250 Hz	boost/cut
LO	+/-15 dB	50 Hz	shelving



14. EFFECT 1

Control for adjusting the EFFECT 1 level. This control is electrically arranged after (post fader) the channel fader (18), so that the signal level depends on the position of the channel fader. Please carefully control the send signal to the integrated effect module. The PEAK indication of the effect module (41) may only light up briefly at dynamic peaks. If the LED lights continuously, the unit is being overdriven.

The EFFECT 1 path can be used to send a signal to a separate external effect unit or to drive a separate monitor power amp.

For further information see: 36-44, EFFECT 1

15. EFFECT 2

Control for sending a signal to the built-in digital effect unit (reverb/delay). This control is also arranged after the channel fader (18); the effect signal level also depends on the position of the channel fader.

Please carefully control the send signal to the integrated effect module. The PEAK indication of the effect module (56) may only light up briefly at dynamic peaks. If the LED lights continuously, the unit is being overdriven.

For further information see: 45-53, EFFECT 2

16. MONITOR

Control for driving a monitor power amp. This control does not depend on the position of the channel fader (18).

17. BAL

This control determines the stereo position of the input signal. In central position the stereo signal is divided equally between both master channels left and right.

18. CHANNEL FADER

The channel fader is used to adjust the volume of the single channel and the volume balance between the individual channels.

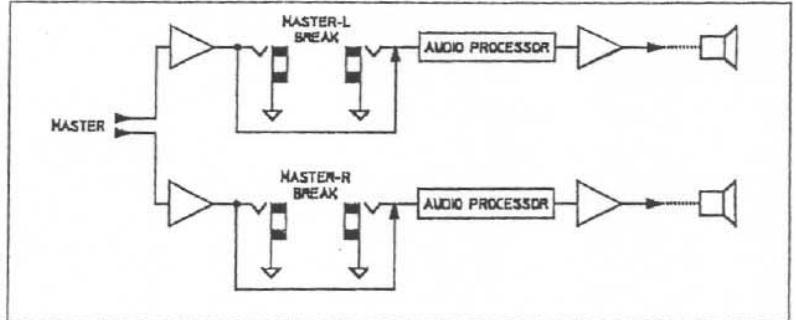
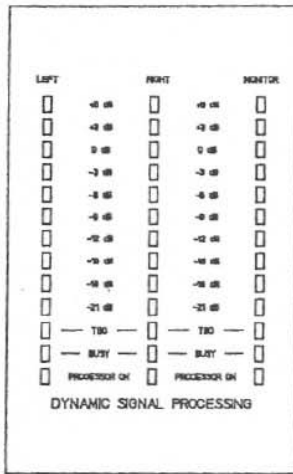
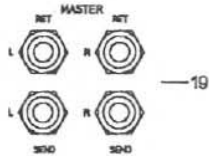
Try to adjust the fader near the 0 dB position. This will give you the possibility to control the volume with sufficient control displacement even if you have great level differences between the different input channels. The master volume of the complete unit is controlled by the master faders.

INPUT / STEREO

MASTER

19. MASTER RET/SEND

These jacks are used for looping in an equalizer etc. into the master signal path. The RETURN jacks interrupt the master signal. The SEND jacks can also be used as master out e.g. for additional power amps.



20. LEFT + RIGHT / MONITOR LED Level Meter

The LED Level Meters show the power modulation of the power amplifiers. The area from +3 dB to +6 dB indicates risk of overdriving.

Please avoid overdriving. The unit or connected loudspeaker cabinets could be damaged.

TBC

The short-term peak output power of the PCA power amps is considerably higher than the rated output power in order to give you excellent dynamic behaviour. The "dynamic headroom" (IHF-A) is 1.5 dB which is equivalent to approximately 350 Watts/4Ohms output power. The TBC circuit contains a simple 1st order voice coil model to simulate the thermal behaviour of a typical woofer. At continuous overdriving or modulation with square wave signals this part of the processor reduces the power output to the rated output (250 W / 4 ohms), to protect the connected loudspeaker system against thermal overload of the woofer's voice coil.

Please note that speakers with less power capability than the rated power cannot be protected completely by the "Thermal Brain Circuit".

BUSY

This indicator lights up if the limiter part of the processor is activated. Continuous lighting of the BUSY LED indicates danger of overdriving the amp and should be avoided by reducing the output volume.

PROCESSOR ON

These LED's indicate that the unit is ready for operation.

21. STANDBY + LED

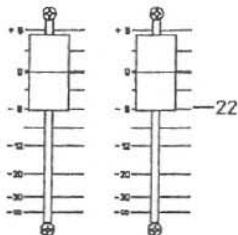
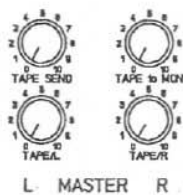
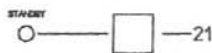
By pressing this button you can mute the outputs MASTER L + R and MONITOR. The red LED will blink and indicates that the complete unit is muted e.g. during breaks.

22. MASTER L + R

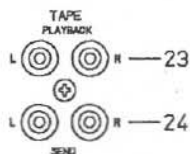
Master volume control for the left and right master output.

For further information see: 56-57. Speaker Outputs

Please make sure that the respective channel fader or at least both master faders are closed before connecting signal sources, to protect your audience from annoying click noise.

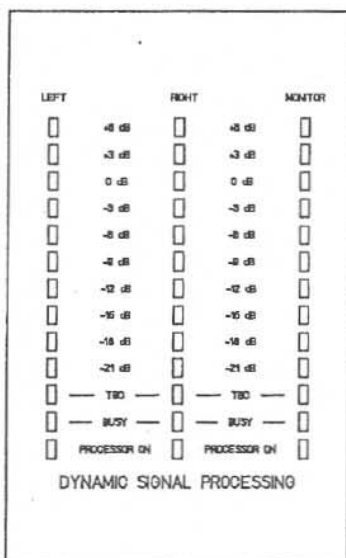


TAPE



23. TAPE PLAYBACK

You can playback a tape or cassette recording via these sockets. The volume is controlled by the TAPE RET control (27) and does not depend on the position of the master faders (22).



24. TAPE SEND

From these sockets you can take the master bus signal for tape recording. The recording level is controlled by the TAPE SEND control (25) and does not depend on the position of the master faders (22).

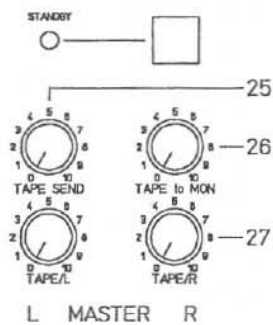
25. TAPE SEND control

With this stereo control knob you can control the output signal of the TAPE SEND sockets (24). This control is for driving a connected tape recorder or cassette player.

26. TAPE to MON

With this control you can add the TAPE signal to the MONITOR section. The volume does not depend on the position of the RET controls (27).

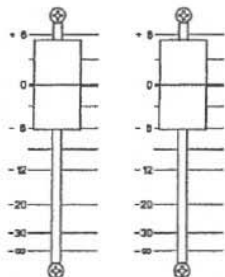
(Very useful for playback performances)



27. TAPE RET control

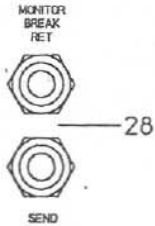
These stereo controls adjust the volume of the tape playback. This tape signal is fed in behind the Master L + R faders (22) and therefore does not depend on the position of the MASTER L + R faders (22).

You can play back tape signals at any volume without altering the master volume by the master faders (22).



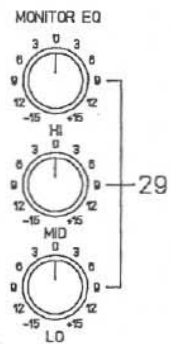
MONITOR

MONITOR



28. MONITOR BREAK RET/SEND

These jacks are used for looping in an equalizer etc. into the monitor signal path. The return jack RET interrupts the bus signal. The SEND jack can also be used as an output e.g. for additional monitor power amps.

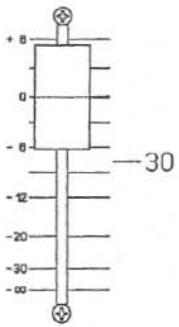


29. EQUALIZING

The EQ section allows for a very comprehensive and effective control of the monitor signal for different frequency regions. Turning the control to the right increases the respective frequency region. Turning left will decrease the respective frequency region.

When adjusting you should always start at a neutral position, i.e. all controls are in central position. If possible, do not use extreme control positions; normally a small sound correction is enough and delivers the best sound quality. Take as orientation the naturalness of the reproduction and depend on your musical trained ear for sound checking. You can effectively avoid acoustical feedback by operating of the mid control (MID) gently. Avoid too much gain especially in this frequency region; a slight lowering allows an amplification of microphone signals with little danger of feedback.

MONITOR



30. MONITOR

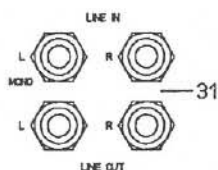
Volume control for the monitor bus output and monitor power amplifier.

For further information see : **57, Speaker Output Monitor**

LINE

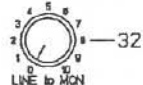
31. LINE IN L/MONO + R

You can feed in a stereo signal, e.g. from submixers via these jacks. The level of this signal depends on the position of the master faders.



32. LINE OUT L + R

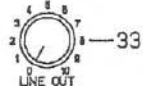
You can take the master bus signal (pre master fader) from these sockets. The LINE OUT signal therefore does not depend on the position of the MASTER L + R faders (25). You can feed via these jacks a separately controllable master bus signal e.g. to a master mixing desk or into an separate amplifier / speaker circuit for monitor purposes.



33. LINE to MON

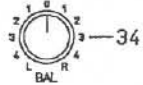
With this control you can add the LINE signal to the MONITOR section.

The volume does not depend on the position of the LINE master fader (35).

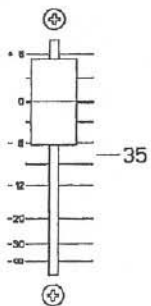


34. LINE OUT control

With this stereo control knob you can adjust the output signal for the LINE OUT jacks (31).



LINE IN



35. LINE IN control

With this stereo control knob you can control the LINE IN signal from the LINE IN sockets, coming from the LINE IN jacks (35) and the mixing in of this signal into the master bus. The master volume depends on the position of the MASTER L + R faders (22).

EFFECT 1

EFFECT 1

36. EXTERN EFFECT/RET

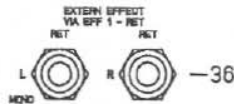
These jacks are used for looping in of external effect units into the EFF 1 path, parallel to the built-in EFF 1.

With mono units you should use the jack RET L/MONO.

The RETURN sockets must be connected with the output of the external effect unit.

37. EXTERN EFFECT/SEND

The EFFECT SEND jack must be connected with the input of the external effect unit.



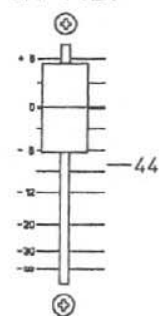
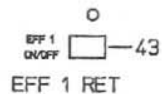
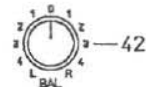
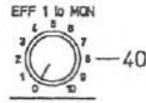
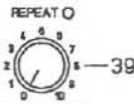
38. PROGRAM SELECT

Program switch for 16 stereo effect programs, of the built-in effect 1 unit.

REVERB
R1 SMALL REV
R2 MEDIUM REV
R3 LARGE REV
E/R1 E/R SMALL
E/R2 E/R MEDIUM
E/R3 E/R LARGE
CH CHORUS
PT PITCH

CHARACTERISTICS:

SHORT REVERB TIME, SMALL ROOM
 MEDIUM REVERB TIME, MEDIUM HALL
 LARGE REVERB TIME, LARGE HALL
 ECHO/REVERB, SMALL ROOM
 ECHO/REVERB, MEDIUM HALL
 ECHO/REVERB, LARGE HALL
 FOR VOCALS, BRASS, WOODWINDS
 DUAL PITCH SHIFTER



DELAY
D1 DELAY 60 ms
D2 DELAY 100 ms
D3 DELAY 170 ms
D4 DELAY 280 ms
D5 DELAY 460 ms
D6 L-R DLY 170 ms
D7 L-R DLY 280 ms
D8 L-R DLY 460 ms

39. REPEAT + LED

With this control the amount of echo repeats will be adjusted at the delay program. The green LED indicates that this control is active.

40. EFF 1 to MON

With this control the reverb or echo signal adjusted in EFF 1 can be mixed to the monitor channel.

41. EFF 1 SEND + PEAK LED

With this control you can adjust the input level for the built-in effect unit and the output level on the EXTERN EFFECT SEND jack (37). The PEAK LED indicates risk of overdriving the built-in effect unit. Please adjust the control EFF SEND so that the LED only lights up briefly at dynamic signal peaks.

42. BAL

This control determines the stereo position of the effect signal. In central position the stereo signal is divided equally between both master channels left and right. Please note that e.g. the right channel of a fed-in stereo signal will be attenuated, if the BAL control is turned to the left (counterclockwise). The same happens for the left channel if the BAL control is turned to the right (clockwise).

43. EFF 1 ON/OFF + LED

If the button is pushed (green LED lights up), the built-in effect module is switched on. The switch does not affect the signal of external effect units, which are connected to the jacks (36, 37). In order to switch EFF 1 with a foot switch, EFF 1 must be switched on with the button (green LED lights up). The LED changes with the switching status of the foot switch.

44. EFF 1 RET

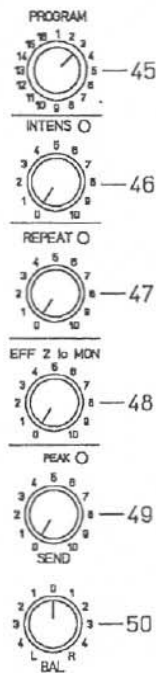
Stereo fader for mixing in the effect 1 signal and/or the external effect signal to the master signal.

EFFECT 2

45. PROGRAM

Program switch for 16 stereo effect programs (echo effect), of the built-in effect 2 unit.

STEREO-DELAY		STEREO-EFFECTS	
1	DELAY 100 ms	11	GUITAR FLANGER 1 *
2	DELAY 145 ms	12	GUITAR FLANGER 2 *
3	DELAY 190 ms	13	GUITAR CHORUS 1
4	DELAY 235 ms	14	GUITAR CHORUS 2
5	DELAY 280 ms	15	VOICE DOUBLING 1
6	DELAY 325 ms	16	VOICE DOUBLING 2
7	DELAY 370 ms		
8	DELAY 415 ms		
9	DELAY 460 ms		
10	DELAY 512 ms		



NOTE

The effects, marked with an asterisk (*), are especially effective with this setting of the INTENS and REPEAT controls.



46. INTENS + LED

With this control the effect volume can be adjusted with the delay or effect programs. The green LED indicates at which program this control is active.

47. REPEAT + LED

With this control the amount of echo repeats will be adjusted at the delay programs and the feedback at flanger effect. The green LED indicates at which programs this control is active.

48. EFF 2 to MON

With this control the effect signal which was adjusted in EFF 2 can be mixed to the monitor channel.

49. EFF SEND + PEAK LED

With this control you can set the input level for the built-in effect unit.

The PEAK LED indicates risk of overdriving the effect unit. Please adjust the control EFF SEND so that the LED only lights up briefly at dynamic signal peaks.

50. BAL

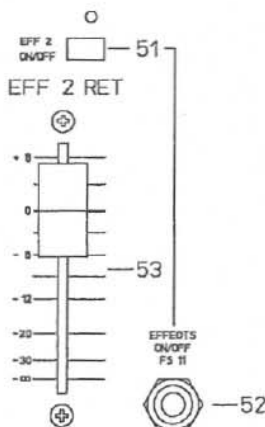
This control determines the stereo position of the effect signal. In central position the stereo signal is divided equally between both master channels left and right.

51. EFF 2 ON/OFF + LED

If the button is pushed (green LED lights up), the effect module is switched on.

52. EFFECT ON/OFF FS-11

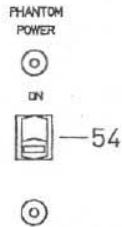
Both effect units can be switched on and off by an optional foot switch FS-11. Both effects must be on and a foot switch FS-11 has to be connected to the jack (52). The red LED in the foot switch lights up if the effect is ON. The LED's EFF ON/OFF show also the status of the foot switch at operation.



63. EFF 2 RET

Stereo fader for mixing in the effect 2 signal to the master signal.

MASTER



MASTER

54. PHANTOM POWER

Central switch for the 48 V phantom power supply for the MIC sockets (1-9).

If you are using phantom powered microphones (e.g. condenser microphones), they can be supplied by the PSX power supply. Separate batteries for the microphones are not necessary.

Please only switch the phantom power supply on and off if the PSX 1250 is switched off.

With PHANTOM POWER ON you must not connect unbalanced signal sources (keyboards, mixers) to the XLR sockets. These units could be damaged or destroyed by the phantom voltage.

ATTENTION! Important Note!

Basically speaking, phantom-powered microphones and balanced dynamic microphones can be operated simultaneously.

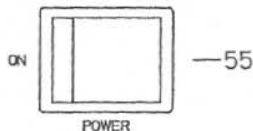
However, there are some balanced dynamic microphones, which are especially sensitive and could possibly be damaged by the phantom voltage. Please read the operating manuals of your microphones carefully.

For reasons of safety, always ensure that the PSX 1250 is switched off (55. POWER), if balanced dynamic microphones are to be connected to the mic input sockets. You will thus avoid possible damage of these especially sensitive dynamic microphones.

55. POWER

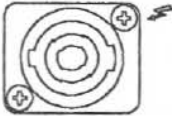
Mains switch for switching the unit on and off.

The unit is ready for operation if both PROCESSOR ON LEDs (20) are lit and the power relays have switched the output stages to the speaker outputs.

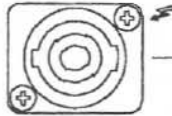


Make sure that both master volume faders are closed when you switch the unit on. You will protect your audience and your equipment from unnecessary inconvenience caused by unwanted amplification and possible feedback.

RIGHT
SPEAKER OUTPUT
RATED Z 4 OHMS
RATED POWER 250W



LEFT
SPEAKER OUTPUT
RATED Z 4 OHMS
RATED POWER 250W



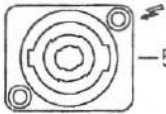
56

56. SPEAKER OUTPUT RIGHT + LEFT

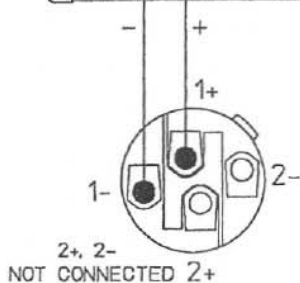
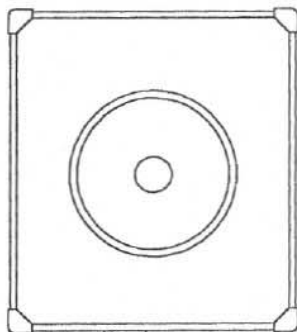
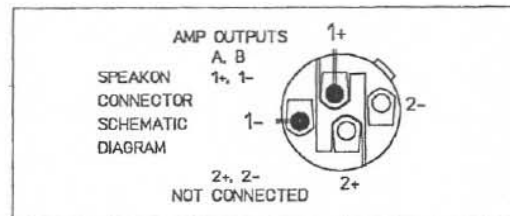
57. SPEAKER OUTPUT MONITOR

The PSX power mixers are supplied with professional SPEAKON high power connectors. This mechanically and electrically safe connection complies with all safety demands and regulations and allows the use of high-powered loudspeaker cables up to a cross-sectional area of $4 \times 2,5 \text{ mm}^2$.

MONITOR
SPEAKER OUTPUT
RATED Z 2 OHMS
RATED POWER 200W



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Check the polarity of the connected speaker cabinets

To prevent acoustic cancellation problems, the polarity of the loudspeaker cabinets connected to the outputs must be equal. Otherwise the bass can sound muddy, suppressed and unprecise and weird midrange lobing problems can occur.

A very simple checking method involves a 9 V battery. If the + pole of the battery is connected to the + pole of the loudspeaker connector, the cone of the woofer should move outwards.

The correct polarity of mid-range and high-range speakers cannot be checked in this way, because some crossover networks change the polarity of speakers in the mid-range and high-range region.

E-V loudspeaker cabinets are internally wired correctly and do not need any polarity check procedure.

SPECIFICATIONS

SPECIFICATIONS

Standard specifications: IEC 268 part 3

IHF-A

0dB = 1V (RMS)

A. POWER SUPPLY

1. Mains:	AC
2. Rated voltage:	120 V
3. Rated frequency:	50 - 60 Hz
4. Rated power consumption:	1650 watts
5. Normal power consumption:	520 watts
6. Tolerance of mains voltage:	± 10 %

B. INPUT CHARACTERISTICS

Input sockets	Rated Input Level *1	Max. Input Level
MIC	-56dB (1.5mV)	-2dB (780mV)
LINE (Mono)	-38dB (13mV)	+18dB (7.6V)
LINE (L + R)	-38dB (13mV)	+18dB (7.6V)
TAPE-PLAYBACK (L+R)	-14dB (200mV)	+12dB (4.1V)
LINE-IN/MASTER (L+R)	-10dB (300mV)	+11dB (3.4V)
EFF1-RETURN (L + R)	-4dB (600mV)	+10dB (3.0V)
MASTER BREAK/RETURN	0dB (1.0V)	n.a.
MONITOR BREAK/RETURN	+1dB (1.1V)	n.a.

C. OUTPUT CHARACTERISTICS

Output sockets	Rated Load Impedance	Output Level *2	
		Rated Value	Max. Level before Clipping
SPEAKER/MASTER (L + R)	4 Ohm	250W	n.a.
	8 Ohm	180W	n.a.
SPEAKER/MONITOR	2 Ohm	200W	n.a.
	4 Ohm	160W	n.a.
MASTER BREAK/SEND	10 k Ohm	0dB (1.0V)	[+16dB (6.0V)]
MONITOR BREAK/SEND	10 k Ohm	+1dB (1.1V)	[+17dB (7.5V)]
EFF1 SEND	10 k Ohm	+2dB (1.2V)	+17dB (7.5V)
LINE OUT (L + R)	10 k Ohm	+2dB (1.3V)	+17dB (7.5V)
TAPE SEND (L + R)	47 k Ohm	-2dB (800mV)	+14dB (5.0V)

SPECIFICATIONS

Output Sockets	Stabilizing
SPEAKER/MASTER (L + R)	3 % (0.26dB)
SPEAKER/MONITOR	6.5 % (0.55dB)

SINGLE CHANNEL OUTPUT POWER

(measured with 'Dynamic Headroom'-test signal according to IHF-A: 1 kHz

Tone burst, 20 ms ON, 480 ms OFF, REPEAT 0.5 s)

SPEAKER/MASTER (L or R)	P = 385 W
- DYNAMIC HEADROOM	1.9 dB
SPEAKER/MONITOR	P = 282 W
- DYNAMIC HEADROOM	1.5 dB

D. FREQUENCY RESPONSE

-3 dB referenced to mid-band level

1. MIC – SPEAKER	: 8 Hz - 55 kHz
2. LINE – SPEAKER	: 8 Hz - 25 kHz

E. AMPLITUDE NON-LINEARITIES

1. POWER AMPS / MASTER

(measured from BREAK-RETURN to SPEAKER-OUT)

1.1 Rated Total Harmonic Distortion	k ≤ 0.3 %
1.2 Norm Total Harmonic Distortion	k ≤ 0.03 %

2. POWER AMP / MONITOR

(measured from BREAK-RETURN to SPEAKER-OUT)

2.1 Rated Total Harmonic Distortion	k ≤ 0.4 %
2.2 Norm Total Harmonic Distortion	k ≤ 0.05 %

3. MIXING DESK

(measured at BREAK-SEND)

3.1 Norm Total Harmonic Distortion	k2 < 0.018%
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all higher distortion products lower than measurement limit (measured with spectrum analyzer)

SPECIFICATIONS

F. NOISE LEVEL

- R(Q) = 200 Ohm between pin 2 and pin 3 of the XLR input socket
- U(F) = Noise voltage, unweighted with B = 20Hz ... 20 kHz
- U(G) = Noise voltage, frequency weighting filter according CCIR, quasi peak-weighted (IEC 268-1)
- U(A) = Noise voltage, dB(A) frequency-weighted, RMS (IEC 268-1)
- S/N ratio ref. rated output voltage (power)

1. Rated noise level (typ.)

	Noise-voltage	S/N Ratio	equiv. input noise voltage	equiv. input noise level
1.1 U(F)	50 mV	55 dBq	2.6 μ V	- 112 dB
1.2 U(G)	95 mV	51 dBqp	4.5 μ V	- 107 dB(G)
1.3 U(A)	18 mV	65 dBp	0.86 μ V	- 121 dB(A)

2. Residual output noise

- 2.1 U(F) = 1.7 mV (85 dBq)
- 2.2 U(G) = 3.1 mV (80 dBqp)
- 2.3 U(A) = 0.7 mV (93 dBp)

G. CROSSTALK ATTENUATION

- 1. Input channel to input channel lower than noise voltage
- 2. Stereo channel:
 - R \rightarrow L 63 dB *3
 - L \rightarrow R 63 dB *3

H. DIMENSIONS

- Height : 217 mm
- Width : 740 mm
- Depth : 570 mm

I. WEIGHT

26.1 kg (57.5 lbs)

*1 : All frequency-dependent level controls full open

*2 : All output levels measured with the measurement signal connected to a MIC input

*3 : Input of measured channel shorted

- Subject to modification without notice! -